Amendments to the Claims

Please amend Claims 1, 2-4, 16, 22-25, 32, and 62. The Claim Listing below will replace all prior versions of the claims in the application:

Claim Listing

1. (Currently Amended) In a communications system for transmitting a near end digital signal using a compression code comprising a plurality of parameters including a first parameter, said parameters representing an audio signal comprising a plurality of audio characteristics, said compression code being decodable by a plurality of decoding procedures, said communications system also transmitting a far end digital signal using a compression code, apparatus for reducing echo in said near end digital signal comprising:

a reading unit responsive to said near end digital signal to read at least said first parameter of said plurality of parameters,

a decoder to perform at least one of said plurality of decoding procedures on said near end digital signal and said far end digital signal and generate at least partially decoded near end signals and at least partially decoded far end signals,

responsive to said at least partially decoded near end signals and at least partially decoded far end signals, an adjustment unit to adjust said first parameter to generate an adjusted first parameter,

an echo likelihood estimator to estimate ~~the~~ <u>an</u> echo <u>likelihood</u> in said near end signal <u>as a function of a ratio of powers of the near end signal and the far end signal</u>;

responsive to said echo <u>likelihood</u> estimate, a replacement unit to replace at least said first parameter with said adjusted first parameter in said near end digital signal to reduce echo in the near end digital signal, and

a transmitter to transmit said near end digital signal with reduced echo.

2. (Currently Amended) Apparatus, as claimed in claim 1, wherein said first parameter is a quantized first parameter and wherein said ~~processor~~ <u>adjustment unit</u> generates said

adjusted first parameter in part by quantizing said adjusted first parameter before writing said adjusted first parameter into said near end digital signal.

3. (Currently Amended) Apparatus, as claimed in claim 1, wherein said ~~processor~~ echo likelihood estimator is responsive to said at least partially decoded near end signals and said at least partially decoded far end signals to generate an echo likelihood signal representing the amount of echo present in said partially decoded near end signals, and wherein said ~~processor~~ adjustment unit is responsive to said echo likelihood signal to adjust said first parameter.

4. (Currently Amended) Apparatus, as claimed in claim 3, wherein said characteristics comprise spectral shape and wherein said first parameter comprises a representation of filter coefficients, and wherein said ~~processor~~ adjustment unit is responsive to said echo likelihood signal to adjust said representation of filter coefficients towards a magnitude frequency response.

5. (Original) Apparatus, as claimed in claim 4, wherein said representation of filter coefficients comprises line spectral frequencies.

6. (Original) Apparatus, as claimed in claim 4, wherein said representation of filter coefficients comprises log area ratios.

7. (Original) Apparatus, as claimed in claim 4, wherein said magnitude frequency response corresponds to background noise.

8. (Original) Apparatus, as claimed in claim 1, wherein said characteristics comprise the overall level of said audio signal and wherein said first parameter comprises codebook gain.

9. (Original) Apparatus, as claimed in claim 1, wherein said first parameter comprises a codebook vector parameter.

10. (Original) Apparatus, as claimed in claim 1, wherein said characteristics comprise period of long-term correlation and wherein said first parameter comprises a pitch period parameter.

11. (Original) Apparatus, as claimed in claim 1, wherein said characteristics comprise strength of long-term correlation and wherein said first parameter comprises a pitch gain parameter.

12. (Original) Apparatus, as claimed in claim 1, wherein said characteristics comprise spectral shape and wherein said first parameter comprises a representation of filter coefficients.

13. (Original) Apparatus, as claimed in claim 12, wherein said representation of filter coefficients comprises log area ratios.

14. (Original) Apparatus, as claimed in claim 12, wherein said representation of filter coefficients comprises line spectral frequencies.

15. (Original) Apparatus, as claimed in claim 12, wherein said representation of filter coefficients corresponds to a linear predictive coding synthesis filter.

16. (Currently Amended) Apparatus, as claimed in claim 1, wherein said first parameter corresponds to a first characteristic of said plurality of audio characteristics, wherein said plurality of decoding procedures comprises at least one decoding procedure avoiding substantial altering of said first characteristic and wherein said ~~processor~~ decoder avoids performing said at least one decoding procedure.

17. (Original) Apparatus, as claimed in claim 16, wherein said audio characteristic comprises power and wherein said first characteristic comprises power.

18. (Previously Presented) Apparatus, as claimed in claim 16, wherein said at least one decoding procedure comprises post-filtering.

19. (Original) Apparatus, as claimed in claim 1, wherein said compression code comprises a linear predictive code.

20. (Original) Apparatus, as claimed in claim 1, wherein said compression code comprises regular pulse excitation – long term prediction code.

21. (Original) Apparatus, as claimed in claim 1, wherein said compression code comprises code-excited linear prediction code.

22. (Currently Amended) Apparatus, as claimed in claim 1, wherein said first parameter comprises a series of first parameters received over time, wherein said ~~processor~~ reading unit is responsive to said near end digital signal to read said series of first parameters, and wherein said ~~processor~~ adjustment unit is responsive to said at least partially decoded near end and far end signals and to at least a plurality of said series of first parameters to generate said adjusted first parameter.

23. (Currently Amended) Apparatus, as claimed in claim 1, wherein said compression code is arranged in frames of said digital signals and wherein said frames comprise a plurality of subframes each comprising said first parameter, wherein said ~~processor~~ reading unit is responsive to said compression code to read at least said first parameter from each of said plurality of subframes, and wherein said ~~processor~~ replacement unit replaces said first parameter with said adjusted first parameter in each of said plurality of subframes.

24. (Currently Amended) Apparatus, as claimed in claim 23, wherein said ~~processor~~ reading unit reads said first parameter from a first of said subframes, said decoder begins to perform at least a plurality of said decoding procedures on said near end digital signal during said first subframe and said replacement unit replaces said first parameter with said adjusted first parameter before processing a subframe following the first subframe so as to achieve lower delay.

25. (Currently Amended) Apparatus, as claimed in claim 1, wherein said compression code is arranged in frames of said digital signals and wherein said frames comprise a plurality of subframes each comprising said first parameter, wherein said ~~processor~~ decoder performs at least a plurality of said decoding procedures during a first of said subframes to generate said at least partially decoded near end and far end signals, said reading unit reads said first parameter from a second of said subframes occurring subsequent to said first subframe, said adjustment unit generates said adjusted first parameter in response to said at least partially decoded near end and far end signals and said first parameter, and said replacement unit replaces said first parameter of said second subframe with said adjusted first parameter.

26. (Previously Presented) In a communications system for transmitting a near end digital signal comprising code samples, said code samples comprising first bits using a compression code and second bits using a linear code, said code samples representing an audio signal, said audio signal having a plurality of audio characteristics, said system also transmitting a far end digital signal, apparatus for reducing echo comprising:

a processor responsive to said near end digital signal and said far end digital signal to adjust said first bits and said second bits , without decoding said compression code in said near end digital signal, to reduce echo in the near end digital signal; and

a transmitter to transmit the first and second bits in an adjusted state to a far end device to present the first and second bits in an audible form to an end user.

27. (Canceled)

28. (Original) Apparatus, as claimed in claim 26, wherein said linear code comprises pulse code modulation (PCM) code.

29. (Original) Apparatus, as claimed in claim 26, wherein said compression code samples conform to the tandem-free operation of the global system for mobile communications standard.

30. (Original) Apparatus, as claimed in claim 26, wherein said first bits comprise the two least significant bits of said samples and wherein said second bits comprise the 6 most significant bits of said samples.

31. (Original) Apparatus, as claimed in claim 29, wherein said 6 most significant bits comprise PCM code.

32. (Currently Amended) In a communications system for transmitting a near end digital signal using a compression code comprising a plurality of parameters including a first parameter, said parameters representing an audio signal comprising a plurality of audio characteristics, said compression code being decodable by a plurality of decoding procedures, said communications system also transmitting a far end digital signal using a compression code, a method of reducing echo in-said near end digital signal comprising:

reading at least said first parameter of said plurality of parameters in response to said near end digital signal;

performing at least one of said plurality of decoding procedures on said near end digital signal and said far end digital signal to generate at least partially decoded near end signals and at least partially decoded far end signals;

adjusting said first parameter in response to said at least partially decoded near end signals and at least partially decoded far end signals to generate an adjusted first parameter;

estimating ~~the~~ an echo likelihood in said near end signal as a function of a ratio of powers of the near end signal and the far end signal;

replacing at least said first parameter with said adjusted first parameter in said near end digital signal, in response to said echo likelihood estimate, to reduce echo in the near end digital signal, and

transmitting said near end digital signal with reduced echo.

33. (Previously Presented) A method, as claimed in claim 32, wherein said first parameter is a quantized first parameter and wherein said adjusting comprises generating said adjusted first parameter in part by quantizing said adjusted first parameter.

34. (Previously Presented) A method, as claimed in claim 32, wherein said adjusting comprises generating an echo likelihood signal representing the amount of echo present in said partially decoded near end signals in response to said at least partially decoded near end signals and said at least partially decoded far end signals, and wherein said adjusting further comprises adjusting said first parameter in response to said echo likelihood signal.

35. (Previously Presented) A method, as claimed in claim32, wherein said characteristics comprise spectral shape and wherein said first parameter comprises a representation of filter coefficients, and wherein said adjusting comprises adjusting said representation of filter coefficients towards a magnitude frequency response in response to said echo likelihood signal.

36. (Previously Presented) A method, as claimed in claim 35, wherein said representation of filter coefficients comprises line spectral frequencies.

37. (Previously Presented) A method, as claimed in claim 35, wherein said representation of filter coefficients comprises log area ratios.

38. (Previously Presented) A method, as claimed in claim 35, wherein said magnitude frequency response corresponds to background noise.

39. (Previously Presented) A method, as claimed in claim 32, wherein said characteristics comprise the overall level of said audio signal and wherein said first parameter comprises codebook gain.

40. (Previously Presented) A method, as claimed in claim 32, wherein said first parameter comprises a codebook vector parameter.

41. (Previously Presented) A method, as claimed in claim 32, wherein said characteristics comprise period of long-term correlation and wherein said first parameter comprises a pitch period parameter.

42. (Previously Presented) A method, as claimed in claim 32, wherein said characteristics comprise strength of long-term correlation and wherein said first parameter comprises a pitch gain parameter.

43. (Previously Presented) A method, as claimed in claim 32, wherein said characteristics comprise spectral shape and wherein said first parameter comprises a representation of filter coefficients.

44. (Previously Presented) A method, as claimed in claim 43, wherein said representation of filter coefficients comprises log area ratios.

45. (Previously Presented) A method, as claimed in claim 43, wherein said representation of filter coefficients comprises line spectral frequencies.

46. (Previously Presented) A method, as claimed in claim 43, wherein said representation of filter coefficients corresponds to a linear predictive coding synthesis filter.

47. (Previously Presented) A method, as claimed in claim 32, wherein said first parameter corresponds to a first characteristic of said plurality of audio characteristics, wherein said plurality of decoding procedures comprises at least one decoding procedure avoiding substantial altering of said first characteristic and wherein said performing at least a plurality of said decoding procedures comprises avoiding performing said at least one decoding procedure.

48. (Previously Presented) A method, as claimed in claim 47, wherein said audio characteristic comprises power and wherein said first characteristic comprises power.

49. (Previously Presented) A method, as claimed in claim 47, wherein said at least one decoding procedure comprises post-filtering.

50. (Previously Presented) A method, as claimed in claim 32, wherein said compression code comprises a linear predictive code.

51. (Previously Presented) A method, as claimed in claim 32, wherein said compression code comprises regular pulse excitation – long term prediction code.

52. (Previously Presented) A method, as claimed in claim 32, wherein said compression code comprises code-excited linear prediction code.

53. (Previously Presented) A method, as claimed in claim 32, wherein said first parameter comprises a series of first parameters received over time, wherein said reading comprises reading said series of first parameters, and wherein said adjusting comprises generating said adjusted first parameter in response to said at least partially decoded near end and far end signals and to at least a plurality of said series of first parameters.

54. (Previously Presented) A method, as claimed in claim 32, wherein said compression code is arranged in frames of said digital signals and wherein said frames comprise a plurality of

subframes each comprising said first parameter, wherein said reading comprises reading at least said first parameter from each of said plurality of subframes in response to said compression code, and wherein said replacing comprises replacing said first parameter with said adjusted first parameter in each of said plurality of subframes.

55. (Previously Presented) A method, as claimed in claim 32, wherein said reading comprises reading said first parameter from a first of said subframes, wherein said performing comprises beginning to perform at least a plurality of said decoding procedures on said near end digital signal during said first subframe and wherein said replacing comprises replacing said first parameter with said adjusted first parameter before processing a subframe following the first subframe so as to achieve lower delay.

56. (Previously Presented) A method, as claimed in claim 32, wherein said compression code is arranged in frames of said digital signals and wherein said frames comprise a plurality of subframes each comprising said first parameter, wherein said performing comprises performing at least a plurality of said decoding procedures during a first of said subframes to generate said at least partially decoded near end and far end signals, wherein said reading comprises reading said first parameter from a second of said subframes occurring subsequent to said first subframe, wherein said adjusting comprises generating said adjusted first parameter in response to said at least partially decoded near end and far end signals and said first parameter, and wherein said replacing comprises replacing said first parameter of said second subframe with said adjusted first parameter.

57. (Previously Presented) In a communications system for transmitting a near end digital signal comprising code samples, said code samples comprising first bits using a compression code and second bits using a linear code, said code samples representing an audio signal, said audio signal having a plurality of audio characteristics, said system also transmitting a far end digital signal, a method of reducing echo in said near end digital signal, comprising:

adjusting said first bits and said second bits, without decoding said compression code in said near end digital signal, in response to said near end digital signal

and said far end digital signal to reduce echo characteristics of said near-end digital signal; and

transmitting the first and second bits in an adjusted state to a far end device to present the first and second bits in audible form to an end user.

58. (Previously Presented) A method, as claimed in claim 57, wherein said linear code comprises pulse code modulation (PCM) code.

59. (Previously Presented) A method, as claimed in claim 57, wherein said compression code samples conform to the tandem-free operation of the global system for mobile communications standard.

60. (Previously Presented) A method, as claimed in claim 57, wherein said first bits comprise the two least significant bits of said samples and wherein said second bits comprise the 6 most significant bits of said samples.

61. (Previously Presented) A method, as claimed in claim 60, wherein said 6 most significant bits comprise PCM code.

62. (Currently Amended) Apparatus for reducing echo in a coded domain signal, comprising:

a near end partial decoder to at least partially decode coded near end digital signals, including at least a first parameter of a plurality of parameters representing respective near end audio signals in the coded near end digital signals to form at least partially decoded near end signals;

a far end partial decoder to at least partially decode coded far end digital signals, including at least a first parameter of a plurality of parameters representing respective far end audio signals in the coded far end digital signals to form at least partially decoded far end signals;

a processor responsive to said near end digital signals to read at least said first parameter of first said plurality of parameters in the coded near end digital signals and at

least partially decode said near end digital signal, to read a coded far end digital signal to generate at least partially decoded far end signals and at least partially decoded far end signals, and to estimate an echo likelihood in the near-end signal as a function of a ratio of powers of the near end signal and the far end signal, and responsive to at least said partially decoded near end signals, at least partially decoded far end signals, and said estimated echo likelihood to adjust said first parameter to generate an adjusted first parameter and to replace at least said first parameter with said adjusted first parameter in said near end digital signal to reduce echo in the near end digital signal.